[0063] The present invention has been described in particular detail with respect to one possible embodiment. Those of skill in the art will appreciate that the invention may be practiced in other embodiments. First, the particular naming of the components, signals, capitalization of terms, the attributes, data structures, or any other programming or structural aspect is not mandatory or significant, and the mechanisms that implement the invention or its features may have different names, formats, or protocols. Further, the system may be implemented in a digital signal processor, as described, or entirely in discrete analog elements. Also, the particular division of functionality between the various components described herein is merely exemplary, and not mandatory; functions performed by a single component may instead be performed by multiple components, and functions performed by multiple components may instead performed by a single component.

[0064] Some portions of the above description present the features of the present invention in terms of algorithms and symbolic representations of operations on various signals. These algorithmic descriptions and representations are the means used by those skilled in the data processing arts to most effectively convey the substance of their work to others skilled in the art. These operations, while described functionally or logically, are understood to be implemented by any variety of suitable structures, whether firmware, software, hardware, or a combination thereof. Furthermore, it has also proven convenient at times, to refer to these arrangements of operations as modules, code devices or elements, without loss of generality.

CLAIMS

1. A method of limiting a signal level in a system having one or more predetermined time and frequency domain dependent limits, comprising:

predicting, at each of a plurality of frequencies, an excess amount by which a sound pressure level that would be output by an acoustic transducer in response to an

unattenuated input signal, would exceed at least one of the one or more predetermined time and frequency domain limits;

tracking an instantaneous maximum of the excess predicted sound pressure level;

calculating an average acoustic energy associated with the excess predicted sound pressure level;

calculating at least one attenuation factor based on the average acoustic energy associated with the excess predicted sound pressure level and the instantaneous maximum of the excess predicted sound pressure level; and

applying the at least one attenuation factor to an unattenuated input signal.

2. The method of claim 1, wherein predicting the excess sound pressure level further comprises;

receiving an unattenuated input signal;

determining a model filter transfer function;

applying a filter design technique to the model filter transfer function in order to determine non-quantized numerical filter coefficients which describe a time-domain filter; and

applying the unattenuated input signal to a time-domain filter described by the non-quantized numerical filter coefficients.

3. The method of claim 2, wherein determining the model filter transfer function further comprises: determining the model transfer function from the frequency domain characteristics of an acoustic transducer, a digital-to-analog converter,, and one or more predetermined limits.

4. The method of claim 2 wherein the filter design technique applied to the model filter transfer function is the Yule-Walker method.

5. The method of claim 2 further comprising applying a second transformation to the non-quantized numerical coefficients and quantizing and reformatting the output to produce second order, quantized numerical coefficients of a fixed word length which are compatible with the input requirements of a digital signal processing chip.

6. The method of claim 5 wherein quantizing and reformatting the second order, non-quantized numerical coefficients further comprises:

reformatting the non-quantized numerical coefficients from floating-point numbers to fixed-point numbers;

truncating the fixed point numbers to a pre-determined number of bits; and

formatting the resulting fixed point numbers to make them compatible with the input requirements of a particular signal processing circuit.

7. The method of claim 1 wherein attenuating the unattenuated input signal further comprises:

delaying the unattenuated input signal;

buffering a plurality of attenuation factors; and

synchronizing the delayed, unattenuated input signal with the buffered attenuation factors so as to minimize instantaneous changes to the input signal and reduce undesirable acoustic artifacts that could result from instantaneous changes to the input signal.

8. The method of claim 7 wherein buffering the plurality of attenuation factors further comprises:

determining a target attenuation factor;

comparing the target attenuation factor to a reference attenuation factor; and

populating a buffer with a plurality of attenuation factors determined by interpolating

between the target attenuation factor and the reference attenuation factor.

9. The method of claim 8 wherein, in reference to a digital signal processing system, the reference attenuation factor used for the current time step is the attenuation factor that was applied to the unattenuated, delayed input signal in the previous time step.

10. The method of claim 8, wherein linear interpolation is used to populate the buffer with a plurality of attenuation factors.

11. The method of claim 8 wherein polynomial curve fit interpolation is used to populate the buffer with a plurality of attenuation factors.

12. The method of claim 8 wherein exponential curve fit interpolation is used to populate the buffer with a plurality of attenuation factors.

13. The method of claim 1 wherein determining the average acoustic energy associated with each of a plurality of frequency bands of the excess predicted sound pressure level further comprises approximating, for each frequency band, the time-weighted average of the excess acoustic energy associated with each frequency band.

14. The method of claim 1 wherein determining the attenuation factor further comprises:

determining a first attenuation as a function of the amount by which the average acoustic energy associated with each frequency band exceeds a first predetermined limit, and the amount of time for which the average acoustic energy associated with any frequency band exceeds the predetermined limit;

determining a second attenuation factor as a function of the amount by which the average acoustic energy associated with each frequency band exceeds a second predetermined limit;

determining a third attenuation factor as a function of the amount by which the instantaneous maximum amplitude of the predicted sound pressure level exceeds a third predetermined limit; and

determining a final attenuation factor as a function of the first attenuation factor, the second attenuation factor and the third attenuation factor.

15. The method of claim 1 wherein the sound pressure that is predicted is not normalized based on predetermined limits such that the result is the sound pressure level that would be output by acoustic transducers in response to the unattenuated input signal.

16. The method of claim 14 wherein the final attenuation factor is selected so as to minimize the amount of attenuation applied to the unattenuated input signal.

17. An apparatus comprising;

a first circuit that separates an input electrical representation of a predicted sound pressure level into a predicted sound pressure level in each of a plurality of frequency bands;

a second circuit communicatively coupled to the first circuit that calculates the average acoustic energy associated with the predicted sound pressure level in each frequency band; and

a third circuit coupled to the second circuit which determines an attenuation factor based on the amount by which the average acoustic energy associated with each frequency band exceeds predetermined limits.

18. A signal level limiting apparatus comprising:

a system modeling filter which predicts at each of a plurality of frequencies an excess amount by which a sound pressure level of an acoustic signal, that would be

output by an acoustic transducer in response to an unattenuated input signal, would exceed one or more predetermined limits at each of the plurality of frequencies;

a peak detector communicatively coupled to the system modeling filter which tracks an instantaneous maximum level from the predicted sound pressure level received from the system modeling filter;

an energy detector which calculates an average acoustic energy associated with the excess predicted sound pressure level;

a gain logic block communicatively coupled to the energy detector and the peak detector, which calculates an attenuation factor from the average acoustic energy and the instantaneous maximum levels; and

a gain block communicatively coupled to the gain logic block which applies the attenuation factor to the unattenuated input signal.

19. A signal level limiting apparatus comprising:

a spectral energy detector which measures signal energy associated with an unattenuated input signal in a plurality of frequency bands;

a peak detector which tracks one or more instantaneous maximum amplitudes associated with the signal energy;

an average energy detector which calculates an average acoustic energy associated with the signal energy;

a gain logic block communicatively coupled to the spectral energy detector and the peak detector, which calculates an attenuation factor associated with the average acoustic energy and the one or more instantaneous maximum amplitudes; and

a gain block communicatively coupled to the gain logic block, which applies the attenuation factor to the unattenuated input signal.

20. The apparatus of claim 18, further comprising, a delay buffer which delays the unattenuated input signal by a predetermined amount of time.

21. The apparatus of claim 19, further comprising, a delay buffer, which delays the unattenuated input signal by a predetermined amount of time.

22. The apparatus of claim 18 wherein the energy detector is a spectral energy detector that measures the signal energy associated with an unattenuated input signal in a plurality of frequency bands.

23. The apparatus of claim 22 wherein the spectral energy detector further comprises a bandpass filter bank communicatively coupled to the system modeling filter, wherein the bandpass filter bank separates the excess predicted sound pressure level into a plurality of frequency bands.

24. The apparatus of claim 20, further comprising a gain smoother communicatively coupled to the gain logic block and synchronized to the delay buffer, wherein the gain smoother reduces undesirable acoustic artifacts associated with abruptly changing the attenuation factor to be applied to the unattenuated input signal.

25. The apparatus of claim 21, further comprising a gain smoother communicatively coupled to the gain logic block and synchronized to the delay buffer, wherein the gain smoother reduces undesirable acoustic artifacts associated with abruptly changing the attenuation factor to be applied to the unattenuated input signal.

26. A method of limiting a signal level, comprising:

measuring signal energy associated with an unattenuated input signal in a plurality of

frequency bands;

tracking one or more instantaneous maximum amplitudes associated with the signal energy;

calculating an average acoustic energy associated with the signal energy;

calculating an attenuation factor associated with the average acoustic energy and the one or more instantaneous maximum amplitudes; and

applying the attenuation factor to the unattenuated input signal.